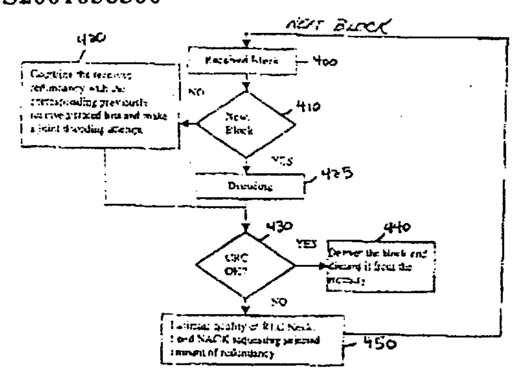
Data communication method and system using an adaptive hybrid-ARQ scheme

Patent number: CN1336049
Publication date: 2002-02-13
Inventor: KHAN F (SE); OLOFSSON H (SE); FURUSKAI A (SE)
Applicant: ERICSSON TELEFON AB L M (SE)
Classification:
- international: H04L1/18
- european: H04L1/16F5; H04L1/16F15; H04L1/18D2
Application number: CN19990814290 19991008
Priority number(s): US19980168064 19981008

Also published as:
WO0021236 (A1)
EP1119936 (A1)
US2001056560 (A1)
CA2346534 (A1)

Abstract not available for CN1336049
Abstract of corresponding document: **US2001056560**

Hybrid ARQ techniques for error handling are described. The amount of redundancy transmitted in response to a first NACK message associated with a first attempt to decode a data block is variable. The number of redundancy units transmitted (and/or requested) can be selected based on various criteria including, for example, estimated channel quality, estimated block quality, memory usage, a number of outstanding blocks, etc.



D2 — CN

[19]中华人民共和国国家知识产权局

[51]Int. Cl⁷
H04L 1/18

[12] 发明专利申请公开说明书

48

[21] 申请号 99814290.5

[43]公开日 2002年2月13日

[11]公开号 CN 1336049A

[22]申请日 1999.10.8 [21]申请号 99814290.5
[30]优先权
[32]1998.10.8 [33]US [31]09/168,064
[86]国际申请 PCT/SE99/01809 1999.10.8
[87]国际公布 WO00/21236 英 2000.4.13
[85]进入国家阶段日期 2001.6.8

[71]申请人 艾利森电话股份有限公司
地址 瑞典斯德哥尔摩
[72]发明人 F·坎 H·奥洛夫松 A·菲吕斯凯
M·赫克

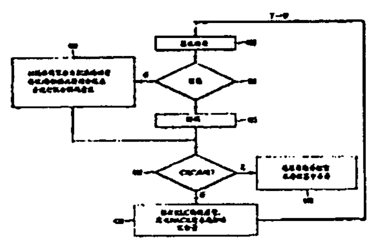
[74]专利代理机构 中国专利代理(香港)有限公司
代理人 程天正 陈景峻

权利要求书4页 说明书10页 附图页数7页

[54]发明名称 使用自适应混合－ARQ方案的数据通信方法和系统

[57]摘要

公开了用于错误处理的混合ARQ技术。响应于与第一次尝试解码数据块相关联的第一个NACK消息而传送的冗余量是可变的。传送(和/或请求)的冗余单元的数目能根据各种标准选择,所述标准包括,例如,估计的信道质量,估计的块质量,存储器使用率,未确认的块数等等。



ISSN 1008-4274

知识产权出版社出版

01.06.08

权　利　要　求　书

1. 通过通信链路传送信息的方法，包含如下步骤:

通过所述的通信链路接收数据块;

确定所述的接收的数据块和所述的通信链路中至少其中一个的质量等级; 并且

请求与所述的数据块相关联的附加信息的数量，所述的数量根据所述的已确定质量等级选择。

2. 权利要求 1 的方法，其中所述的确定步骤进一步包含步骤:

估计与所述的接收数据块相关联的比特错误率。

3. 权利要求 1 的方法，其中所述的确定步骤进一步包含步骤:

采用在解码过程期间所获得的软信息确定所述的质量等级。

4. 权利要求 1 的方法，其中所述的确定步骤进一步包含步骤:

发送一个标识所述的数据块和所述的附加信息数量的消息。

5. 权利要求 1 的方法，其中进一步包含步骤:

选择多个冗余信息单元，作为所述的附加信息量。

6. 权利要求 5 的方法，其中所述的已选择的冗余信息单元的数目相对于所述的已确定质量等级做相反变化。

7. 权利要求 1 的方法，其中所述确定所述质量等级的步骤进一步包含步骤:

单独根据所述的接收数据块确定所述的质量等级。

8. 权利要求 1 的方法，其中所述确定所述质量等级的步骤进一步包含步骤:

单独根据所述的通信链路的质量确定所述的质量等级。

9. 权利要求 1 的方法，其中所述确定所述质量等级的步骤进一步包含步骤:

根据与所述的接收块相关联的质量信息和与所述的通信链路相关联的质量信息之间的组合确定所述的质量等级。

10. 权利要求 1 的方法，进一步包含步骤:

传送所述请求的附加信息数量。

11. 权利要求 1 的方法，进一步包含步骤:

传送不同于所述的已请求数量的附加信息数量。

12. 权利要求 11 的方法，其中所述传送的附加信息数量和所述

请求的附加信息数量根据下面至少其中之一而不同：存储器使用参数，多个未处理块和资源可用性。

13. 用于通过通信链路传送信息的设备，包含：

用于通过通信链路接收数据块的装置；

用于确定所述的接收的数据块和所述的通信链路中至少其中一个的质量等级的装置；以及

用于请求与所述的数据块相关联的附加信息数量的装置，所述的数量根据所述的确定质量等级选择。

14. 权利要求 13 的设备，其中所述用于确定的装置进一步包含：

用于估计与所述的接收数据块相关联的比特错误率的装置。

15. 权利要求 13 的设备，其中所述用于确定的装置进一步包含：

用于采用在解码过程期间所获得的软信息确定所述的质量等级的装置。

16. 权利要求 13 的设备，其中所述用于请求的装置进一步包含：

用于传送标识所述的数据块和所述的附加信息数量的消息的装置。

17. 权利要求 13 的设备，进一步包含：

用于选择多个冗余信息单元作为所述的附加信息数量的装置。

18. 权利要求 13 的设备，其中所述的已选择冗余信息单元的数目相对于所述的已确定质量等级做相反变化。

19. 权利要求 13 的设备，其中所述用于确定所述质量等级的装置进一步包含：

单独根据所述的接收数据块确定所述的质量等级的装置。

20. 权利要求 13 的设备，其中所述用于确定所述质量等级的装置进一步包含：

单独根据所述的通信链路的质量确定所述的质量等级的装置。

21. 权利要求 13 的设备，其中所述用于确定所述质量等级的装置进一步包含：

根据与所述的接收块相关联的质量信息和与所述的通信链路相关联的质量信息的组合确定所述的质量等级的装置。

22. 权利要求 13 的设备，进一步包含：

用于传送所述请求的附加信息数量的装置。

23. 权利要求 13 的设备，进一步包含：

用于传送不同于所述请求数量的附加信息数量的装置。

24. 权利要求 23 的设备，其中所述传送的附加信息数量与所述请求的附加信息数量根据下面至少之一而不同：存储器使用参数，多个未处理块和资源可用性。

25. 在无线通信系统中用于解码信息块的方法，包含如下步骤：

接收一信息块；

解码所述的块；

对所述的解码块执行错误检测技术；

如果所述的块被确定为已错误接收，则确定一个质量等级；

根据所述的质量等级，选择期望的冗余信息量；

传送所述的期望冗余信息量的请求到发送实体；

接收所述请求的冗余信息量；以及

联合解码所述的信息块和所述的冗余信息。

26. 权利要求 25 的方法，其中所述的执行步骤进一步包含步骤：

在所述的信息块上执行循环冗余校验（CRC）。

27. 权利要求 25 的方法，其中所述的质量等级是所述的接收块的质量。

28. 权利要求 25 的方法，其中所述的质量等级是信道的质量，所述的冗余信息将通过该信道被发送。

29. 在发送实体和接收实体之间用于通信信息的方法，包含如下步骤：

在所述的接收实体估计信道质量；

由所述的接收实体传送一个与所述的信道质量相关联的指示；并且

由所述的发送实体，传送一个信息块加上与所述的信息相关联的冗余量，其中所述的量基于所述的指示。

30. 在第一收发器和第二收发器之间用于传送信息的方法，包含如下步骤：

在所述的第一收发器接收数据块；

估计与其中一个所述的接收块和信道相关联的质量；

由所述的第一收发器传送所述的估计质量到第二收发器；

在所述的第二收发器至少部分基于所述的估计质量确定冗余量；并且

由所述的第二收发器传送所述的冗余量到所述的第一收发器。

31. 在第一收发器和第二收发器之间用于通信信息的方法，包含步骤：

在所述的第一收发器接收数据块；

估计与所述的接收块和信道中至少其中一个相关联的第一质量；

由所述的第一收发器传送与基于所述的估计的第一质量的、已选择的冗余量相关联的指示；

由所述的第二收发器传送所述的已选择冗余量到所述的第一收发器；

在所述的第一接收器估计与所述的接收块和接收的冗余信息及所述的信道质量中至少其中一个相关联的第二质量；并且

传送所述的估计的第二质量的指示到所述的第二收发器。

说　明　书

使用自适应混合-ARQ方案的数据通信方法和系统

背景

本发明总地涉及通信系统领域中的错误处理，并且，更具体的，涉及在数字通信系统中采用自动重传请求(ARQ)和变长冗余的错误处理。

商用通信系统的增长，特别是蜂窝无线电话系统的爆炸性增长，已经迫使系统设计者去探索在不把通信质量减低到客户难以忍受的程度的情况下增加系统容量的方法。达到这些目的的一种技术涉及到把使用模拟调制将数据加到载波上的系统变为使用数字调制把数据加到载波上的系统。

在无线数字通信系统中，标准化的空中接口制定了许多系统参数，包括调制类型，脉冲串格式，通信规程等。例如，欧洲电信标准学会（ETSI）已经制定了全球移动通信系统（GSM）标准，该标准采用时分多址（TDMA）在射频（RF）物理信道或链路上传送控制、话音和数据信息，采用符号速率为271ksps的高斯最小频移键控（GMSK）调制方案。在美国，电信工业协会（TIA）已经发布了许多临时标准，如IS-54和IS-136，这些标准定义了各种版本的数字高级移动电话业务（D-AMPS），TDMA系统采用差分正交相移键控（DQPSK）调制方案用于在RF链路上传输数据。

TDMA系统将可用频率细分为一个或多个RF信道。RF信道进一步划分为多个物理信道，对应于TDMA帧中的时隙。逻辑信道由一个或几个规定了调制和编码的物理信道构成。在这些系统中，移动站与多个分散的基站通过在上行链路和下行链路RF信道上发送和接收数字信息脉冲串进行通信。

数字通信系统采用各种技术来处理错误地接收的信息。通常说来，这些技术包括那些辅助接收器纠正错误地接收的信息的技术，例如，前向纠错（FEC）技术，这些技术还包括那些能使错误接收到的信息重新发送到接收方的技术，如自动重传请求（ARQ）技术。FEC技术包括例如在调制之前的数据卷积或块编码。FEC编码包括使用某个（较大）数目的编码比特表示某个数目的数据比特，因此增加了允

许纠正某些错误的冗余比特。这样，通常用卷积码的编码率，如 1/2 和 1/3 来指示卷积码，其中编码率越低则提供更强的错误保护，但是在给定信道比特率情况下提供较低的用户比特率。

ARQ 技术包含分析接收到的数据块中的错误并且请求重新传送包含错误的数据块。例如，考察图 1 所示的数据块映射例子，图 1 表示根据通用分组无线业务(GPRS)最优化进行运行的无线通信系统，GPRS 最优化已经被推荐为用于 GSM 的分组数据业务。这里，逻辑链路控制（LCC）帧包含一个帧头（FH），一个信息载荷和一个帧检验序列（FCS），逻辑链路控制（LCC）被映射到多个无线链路控制（RLC）块，每个这样的块包含一个块头（BH），一个信息域和一个块检验序列（BCS），RLC 用于由接收方检验信息域的错误。RLC 块进一步映射到物理层脉冲串，即无线信号，其通过 GMSK 调制到载波上来传输。在这个例子中，包含在每个 RLC 块中的信息为了传输能够在四个脉冲串（时隙）上交织。

当被接收方如移动无线电话中的接收器处理时，在解调后，采用块检验序列和公知的循环冗余检验技术，对每个 RLC 块的错误进行评估。如果有错误，则将一个请求发回到发送实体，如在无线通信系统中的基站，指示采用预定的 ARQ 规程重新发送该数据块。

这两个错误控制方案的优点和缺点能通过组合 FEC 和 ARQ 技术得到平衡。这种组合的技术，一般称为混合 ARQ 技术，允许在接收器中采用 FEC 编码方案纠正一些收到的错误，而其它的错误需要重传。正确选择 FEC 编码方案和 ARQ 规程这样就导致一种混合 ARQ 技术，其比采用纯粹 FEC 编码方案的系统具有更高的可靠性，比采用纯粹 ARQ-型错误处理机制的系统具有更大的吞吐量。

一个混合 ARQ 方案的例子能在 GPRS 中发现。GPRS 最优化提供了四个 FEC 编码方案（三个不同编码率的卷积码方案和一个非编码方案）。在四个编码方案之一被选中用于当前的 LLC 帧以后，该帧被分段装入 RLC 块中。如果 RLC 块在接收器中被发现有错误（例如，具有不能纠正的错误）并需要被重传，则起初选择的 FEC 编码方案被用于重传，也就是，该系统使用固定的冗余比特用于重传。在尝试对传送的数据进行成功解码的一个通常被称为软组合的过程中，被重传的数据块可以与较早的已传版本结合起来。

提出的另一个混合 ARQ 方案（有时被称为增量冗余或 I 型混合 ARQ）是：如果起初的已传数据块不能被解码，则该方案提供用于发送的附加冗余比特。该方案的概念性说明见图 2。这里，由接收器进行三种解码尝试。首先，接收器试图解码起初接收的数据块（带有或不带有冗余）。在失败后，接收器接着接收附加的冗余比特 R1，使用该冗余比特结合起初发送的数据块以试图解码。在第三步，接收器获得另一块冗余信息 R2，使用该冗余信息结合起初已接收的数据块和冗余比特块 R1，以试图第三次解码。该过程重复直到成功解码的目的达到为止。

如图 2 所示的技术存在的一个问题是巨大存储器需求，存储器需求大小与直到成功解码为止所需要存储的数据块（以及可能附加的冗余比特块）相关联，数据块的存储是必要的，因为随后的已发送冗余块（如 R1 或 R2）不能被独立解码。由于接收器典型地存储一个与每个接收比特相关联的多比特软值的事实，存储需求加倍，软值表示一个与已接收的比特解码相关联的信用水平。这个问题通过采用由 Samir Kallel 发表的题目为“补充凿孔卷积（CPC）码及其应用”的文章中的技术能够得到部分解决，该文章发表在 1995 年 6 月出版的 IEEE Transcations on Communications，卷 43，第 6 期，第 2005-2009 页。在文章中，作者描述了一种纠错技术，其中每个被重传的数据块本身是独立可解码的以便当存储器空间不可用时，能丢弃以前发送的数据块。

图 2 的方案遇到的第二个问题是大分组的传输时延。这些大的时延的产生是因为，一般地，在成功解码之前，几次冗余重传是需要的。与提出的方案相关的第三个问题是由过时的 ARQ 窗口导致的低效率的带宽利用。由于给定时间的大量未处理数据块（即未确认的数据块）ARQ 窗口是停顿的。

因此，希望提供新的技术来提高 ARQ 方案，减少开销信令，提高存储器利用的效率并以允许更有效处理的方式减少与每个解码相关联的冗余重传的次数。

发明概述

根据本发明，用于传递信息的传统方法和系统所具有的这些和其它缺点和局限得到克服，其中接收器处理已接收的数据块。如果解码

不成功，则作出已接收的信息的质量估计。质量估计能单独基于已经错误接收的具体数据块的质量，单独基于与信道质量相关联的历史数据，或者质量估计能是前面两个的某种组合。例如，质量估计能从在接收器中得到的软值中提取出来。接着，根据质量估计，成功对信息块解码所需要的冗余量被确定下来。接收器接着向发送器发出一个未确认（NACK）消息，确定要进行重传的数据块以及所期望的冗余量，基于此所期望的冗余量被发送。

如果在第二次尝试后解码不成功，则通过确定与初始已发送数据块和随后发送的冗余比特两个都关联的第二次质量估计，该过程继续进行。这个第二次质量估计接着被用于确定请求的冗余信息的下一次的数量，等等。

根据本发明基于测量的混合 ARQ 方案将最小化冗余传输步骤数，从而减少分组传输时延和需求的存储量。这种效果的取得是由于采用基于测量的方案而使成功解码所需求的 ACK/NACK 循环次数减少。本发明的示范实施提供了冗余量的估计，该估计依赖于已接收的以前数据块/冗余数据块的质量和/或信道的质量。然而，本发明的另一个示范实施例也包括这样的情形，即传输的冗余量依赖其它因素，如在给定传输中可用存储量，数据时延需求和/或未处理的（未确认）数据块的数量。例如，当存储量是有限的或要求的时延很严格时，为了增加在下一步成功解码的概率，冗余估计应被放大。

附图详述

通过阅读下面的详细描述并结合附图，本发明的这些和其它目的、特征和优点更加明显，其中：

图 1 说明了在根据 GSM 运行的传统系统中的信息映射；

图 2 说明了传统的变长冗余技术；

图 3（a）是有利地采用本发明的 GSM 通信系统的框图；

图 3（b）是用于说明图 3(a) 中 GSM 系统的示范 GPRS 最优化的框图；

图 4 是一个流程图，说明根据本发明的示范实施例的基于测量的 ARQ 方案；

图 5 是一个表，它说明了被传输的冗余单元数，编码率和相应的码字之间的典型关系；

图 6 表示根据本发明的示范实施例的 ACK/NACK 格式；

图 7(a)说明了采用传统增量冗余方案的数据块传输时间；

图 7(b)说明采用根据本发明的技术传送图 7（a）中的相同数据所花费的图示性数据块传输时间；

图 8 是一个表，说明采用本发明在时延方面的渐增改进。

发明详述

下面的示范实施例在 TDMA 无线通信系统的环境中提出。然而，本领域的技术人员将理解该接入方法仅用于举例说明的目的，并且本发明很容易应用到所有类型的接入方法，包括频分多址（FDMA），TDMA，码分多址（CDMA）和它们的混合。

此外，根据 GSM 通信系统的操作在欧洲电信标准学会（ETSI）文件 ETS 300 573，ETS 300 574 和 ETS 300 578 中得到说明，因此在此引入作为参考。所以，结合提出的分组数据的 GPRS 最优化（今后简单称为“GPRS”）的 GSM 系统的操作在这里仅仅说明到理解本发明所必须的程度。尽管本发明是就增强 GPRS 系统中的示范实施例来描述的，本领域的那些技术人员将理解本发明能被用于很多其它数字通信系统，如那些基于宽带 CDMA 或无线 ATM 的系统等等。

参考图 3（a），根据本发明的典型 GSM 实施例的通信系统 10 被描述出来。系统 10 被设计为分层网络，具有多个管理呼叫的等级。采用一套上行链路和下行链路频率，运行在系统 10 内的移动基站 12 使用在这些频率上分配给它们的时隙参与呼叫。在上一个分层等级上，一组移动交换中心（MSC）14 负责由发起方到目的地的呼叫路由。特别是，这些实体负责建立、控制和终止呼叫。其中一个 MSC 14（被称为网关 MSC）处理与公用交换电话网（PSTN）18 或其它公用和专用网络之间的通信。

在一个较低分层等级上，每个 MSC 14 连接到一组基站控制器（BSC）16。在 GSM 标准之下，BSC 16 在一个被称为 A-接口的标准接口下与 MSC 14 进行通信，该接口是基于 CCITT No. 7 信令系统的移动应用部分。

在一个更低分层等级上，每个 BSC 16 控制一组基地收发器站（BTS）20。每个 BTS 20 包括多个 TRX(没有表示)，它们采用上行链路和下行链路 RF 信道来服务具体的公共地理区域，如一个或多个通

信小区 211。BTS 20 主要提供 RF 链路，用于发送和接收移动站 12 被分配的小区中的去往和来自移动站的数据脉冲串。当用于传送分组数据时，这些信道常常被称为分组数据信道（PDCH）。在一个示范实施例中，多个 BTS 20 被包含在一个无线基站（RBS）22 中。例如，根据一族 RBS-2000 产品，RBS 可以得到配置，RBS-2000 产品由 Telefonaktiebolaget L M Ericsson 即本发明的受让人提供。关于典型移动站 12 和 RBS 22 实现的较多细节，感兴趣的读者可参阅美国专利申请第 08/921,319 号，该专利申请的题目为“一种链路自适应方法--用于采用具有不同符号率的调制方案的链路”，作者为 Magnus Frodigh 等，该专利的公开特别在此引用作为参考。

在蜂窝系统中引入分组数据规程的优点是提供支持高数据速率传输的能力并且同时获得无线接口上的无线频宽的灵活和有效利用。GPRS 的概念是设计用于所谓的“多时隙操作”，即一个单个用户被允许同时占用多个传输资源。

GPRS 的网络结构简图见图 3（b）。由于 GPRS 是 GSM 的优化，许多网络节点/实体与图 3(a)中的描述相似。来自外部网络的信息分组在 GGSN(网关 GPRS 业务节点)100 进入 GPRS 网络。分组接着从 GGSN 经由骨干网 120 被发送到 SGSN(服务 GPRS 支持节点)140，SGSN 140 服务被定址的 GPRS 移动电话所在的区域。分组从 SGSN 140 通过专用的 GPRS 传输被发送到正确的 BSS(基站系统)160。BSS 包括多个基站收发信台（BTS）和一个基站控制器（BSC）200，这里仅有一个 BTS 180 显示出来。BTS 和 BSC 之间的接口被称为 A-bis 接口。BSC 是一个 GSM 规范中的定义，在其它典型系统中术语无线网络控制（RNC）用于具有类似 BSC 功能的节点。分组接着由 BTS 180 通过空中接口发送到使用已选择信息传输速率的远端单元 210。

GPRS 寄存器存放所有的 GPRS 预订数据。GPRS 可以与 GSM 系统的 HLR(归属位置寄存器)220 集成在一起，也可以不与 GSM 系统的 HLR 220 集成在一起。用户数据可以在 SGSN 和 MSC/VLR 240 之间交换以保证业务的互操作，如受限制的漫游业务。如上所述，在 BSC 200 和 MSC/VLR 240 之间的接入网络接口是被称为 A-接口的标准接口，其是基于 CCITT No.7 信令系统的移动应用部分。MSC/VLR 240 也提供经由 PSTN 260 接入到陆地线路系统。

重传技术能在系统 10 中提供，使得接收实体（RBS 180 或 MS210）能请求与来自传输实体（MS 210 或 RBS 180）的 RLC 块相关联的冗余比特。根据本发明的示范实施例，由接收实体请求的和响应于请求（例如，未确认（NACK）消息）传送的冗余比特量是可变的。

更具体地，接收器能估计错误接收的 RLC 块，以获得一些关于接收状况也就是质量如何的估计。该估计能是例如比特错误率和载波--干扰比率（C/I）的测量。接收器接着基于对具体错误接收的 RLC 数据块的质量估计确定来自发送器的请求的冗余量。在下面的讨论中，冗余信息的返回根据冗余单元来描述，其当然能够为任意大小，如一个比特块，一字节或甚至一个单个比特，且能通过采用多项式发生器以已知的方式生成。一般说来，质量估计越低，要求的冗余单元数就越多。除了（或可选地）根据错误已接收数据块本身的质量估计请求冗余量外，根据本发明的示范实施例，用于处理错误的系统和方法也可以考虑信道质量，在该信道上发送数据块发送和已请求的冗余单元。例如，请求的冗余单元的数量可以基于全局质量测量值，如 Q=α*信道质量+（1-α）*已收到的数据块质量，其中 α 是期望的加权值。

于是，根据本发明的典型方法如图 4 的流程图所示。其中，在方框 400 中，接收器接收 RLC 块，RLC 块是数据、以前请求的冗余比特或者是其中的某种组合。如果 RLC 块仅仅包含与以前接收的 RLC 块相关联的冗余比特，则过程沿着"否"的箭头方向从判决框 410 移到方框 420，其中冗余比特与以前接收/存储的相应 RLC 块的比特结合在一起，并且作出联合解码尝试。关于冗余比特如何与较早收到的数据相匹配用于联合解码的详细讨论，有兴趣的读者可参考美国专利申请第 09/131,166 号，题目为“在分组数据无线通信系统中用于块寻址的方法和系统”，该申请于 1998 年 8 月 7 日由 Farooq Khan 等提交，并特别引入作为参考。否则，如果接收的块是新 RLC 块，则过程沿着"是"路径从判决框 410 到方框 425，在方框 425 中新块得到解码。接着过程移动到执行循环冗余校验（CRC）的方框 430。如果 CRC 通过，也就是数据接收正确，则过程移动到方框 440，其中该数据块被递送用于后续处理，例如话音解码等。如果 CRC 不成功，则过程移动到方框 450，在这里作出错误接收块的质量估计，例如基于相关的 BER 或

C/I 参数。质量估计（和可能下面所述的其它因素）接着被用于选择期望的用于下一次解码尝试的冗余比特量。接收器接着发送一个与该（也可能是其它）RLC 块相关联的该 NACK 消息，该 NACK 消息指示接收器希望发送器发送的冗余比特的数量。过程接着环回，以处理下一个接收的数据块。

本领域的技术人员将理解，在实施卷积编码的示范实施例中，请求发送的冗余单元的数目能被认为基本上等价于规定所期望的编码率来用于一个错误接收的具体块。例如，如图 5 中的表所示，一个 RLC 块包含四个“单元”的数据，请求发送从 1-8 之间任意数的冗余单元会导致在解码数据的第二次尝试时的不同的实际编码率。于是，例如，一个错误接收的 RLC 块虽然具有相对高的质量，但可能导致接收器从发送器中请求仅仅一个冗余单元。在另一方面，一个非常差的接收 RLC 块可能导致接收器为该特定的 RLC 请求 8 个冗余单元。估计的 RLC 块的质量和请求的冗余单元的数量之间的特殊关系可以随着系统的不同而不同，且例如，能通过模拟得到优化，以获得如下所述的使每一数据块的解码尝试的数量最少的期望结果。

一旦接收器估计出了接收的 RLC 块的质量并选择了一个期望的冗余量，这些信息将包含在给发送器的报告中。拿图 5 举例，每个不同的能被传送的冗余单元数可以被分配一个不同的编码或比特组合。那么，接收器能发送一个确认/不确认（ACK/NACK）消息，而对于每个最近接收的 RLC 块来确定期望的冗余量（如果有）。例子在图 6 中给出。

其中，包含信息[5(3),6(0),7(5),0(8),1(0),2(0),3(1),4(0)]的 ACK/NACK 消息被表示出来。在前述的符号中，"5(3)"表示对于序列号为 5 的 RLC 块，接收器请求三个冗余单元。对于序列号为 6 的 RLC 块，接收器包含编码 000，表示由于 RLC 块已被正确接收没有冗余信息需要传送。

正如较早提及的，通过调整请求的冗余量以满足接收块的质量要求，申请人期望：与传统技术比较，每个块需要较少的解码次数，这里是对于每个错误接收的块发送同样冗余量的情况下。通过考察图 7(a)，7(b)和 8 这一点将更明显。

其中，可以比较采用传统的增量冗余机制（图 7(a)）的典型块传

送次数和采用基于测量的变量冗余机制的典型块传送次数。在这个单纯的说明性例子中，一个块周期等于 20ms，在由发送实体对 RLC 块的传送和由该发送实体对相应 ACK/NACK 的接收之间的一个往返行程时间（RTT）是 200ms，并且一个错误接收的 RLC 块需要三个单元的冗余信息（例如，编码速率为 4/7）被正确解码。这样，在图 7(a) 中将看到对于该 RLC 块需要四次传送才通过 CRC 校验，其中在每次失败后，发送实体发送一个另外的冗余单元。通过对比，采用本发明，根据该 RLC 块的估计质量接收器能够请求三个冗余单元，所以只需要两次，因此分别将块发送时延从 680ms 减少到 240ms。本领域的那些技术人员将理解，与这两种方法相关联的时延上的实际不同也可以根据其它条件而变化，举例来说随无线信道条件的变化而变化。此外，时延的不同将随着传统技术所采用的冗余传送步骤的数目增加而增加，如图 8 的表所示。应当理解在前面例子中提供的数值仅仅是举例说明，其目的是使得与本发明相关的优点更清楚。

除了减少时延以外，本发明的示范实施例也减少了 ARQ 窗口将停顿的概率并减少存储器需求。这是因为根据本发明的技术通过确保较快的块解码和传送，把未处理块的数目减到最小。通过阻止停顿的 ARQ 窗口产生的条件，可以获得更有效的带宽利用，因为新的 RLC 块不能在停顿条件出现期间传送。

如较早所述，在起初发送的数据块可以包含一些冗余信息，即可以有一些等级的 FEC 编码。这个初始级的 FEC 编码可以由发送实体基于发送实体参照信道质量接收的信息确定下来。例如，一个移动站可以参照信道质量作出估计并把这些估计转送到基站。接着，基站能使用接收的信道估计来选择合适的冗余量，以便随着荷载信息发送到移动站。更适宜地，基站将选择冗余量，该冗余量使得移动站根据给出的信道质量估计在第一次尝试中解码数据块。可是，本领域的技术人员将理解基站可以依赖较早所述的各种当前系统因素选择一个较大或较小的冗余量。

尽管本发明通过参考仅仅几个示范实施例得到了详细说明，但是本领域的技术人员将理解各种修改可以在没有脱离本发明的情况下进行。例如，关于被发送的冗余单元数的信息，举例来说，可通过发送每一块的测量估计质量被隐含地传递回发送器，而不是如在图 6 的

例子中那样被明确地给出。发送器将接着确定回送的合适的冗余单元数。为确定冗余单元的数目，除了接收的质量测量之外，发送器还可能考虑其他因素如资源可用性等等。因此，本发明仅仅通过下面的权利要求得到定义，该权利要求包含其所有的等价物。

说 明 书 附 图

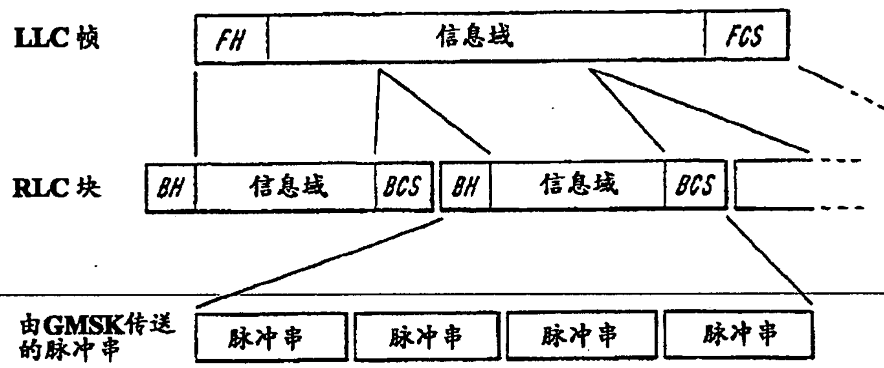


图 1
(现有技术)

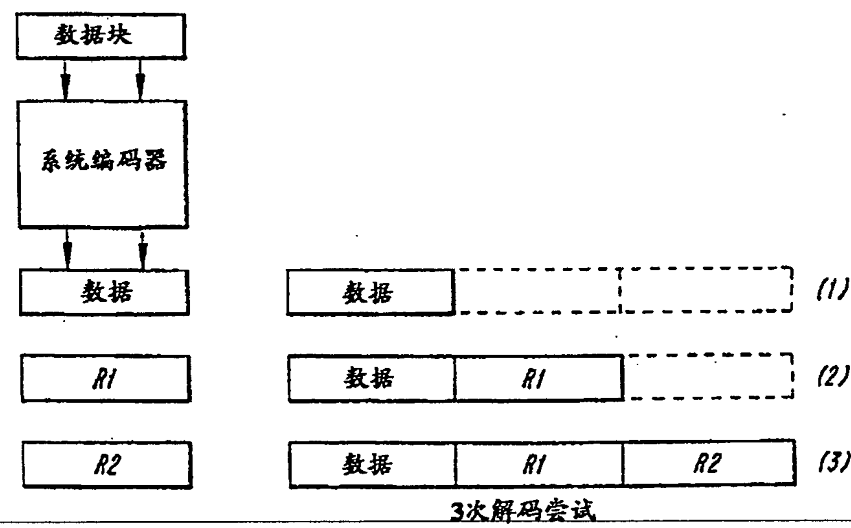


图 2
(现有技术)

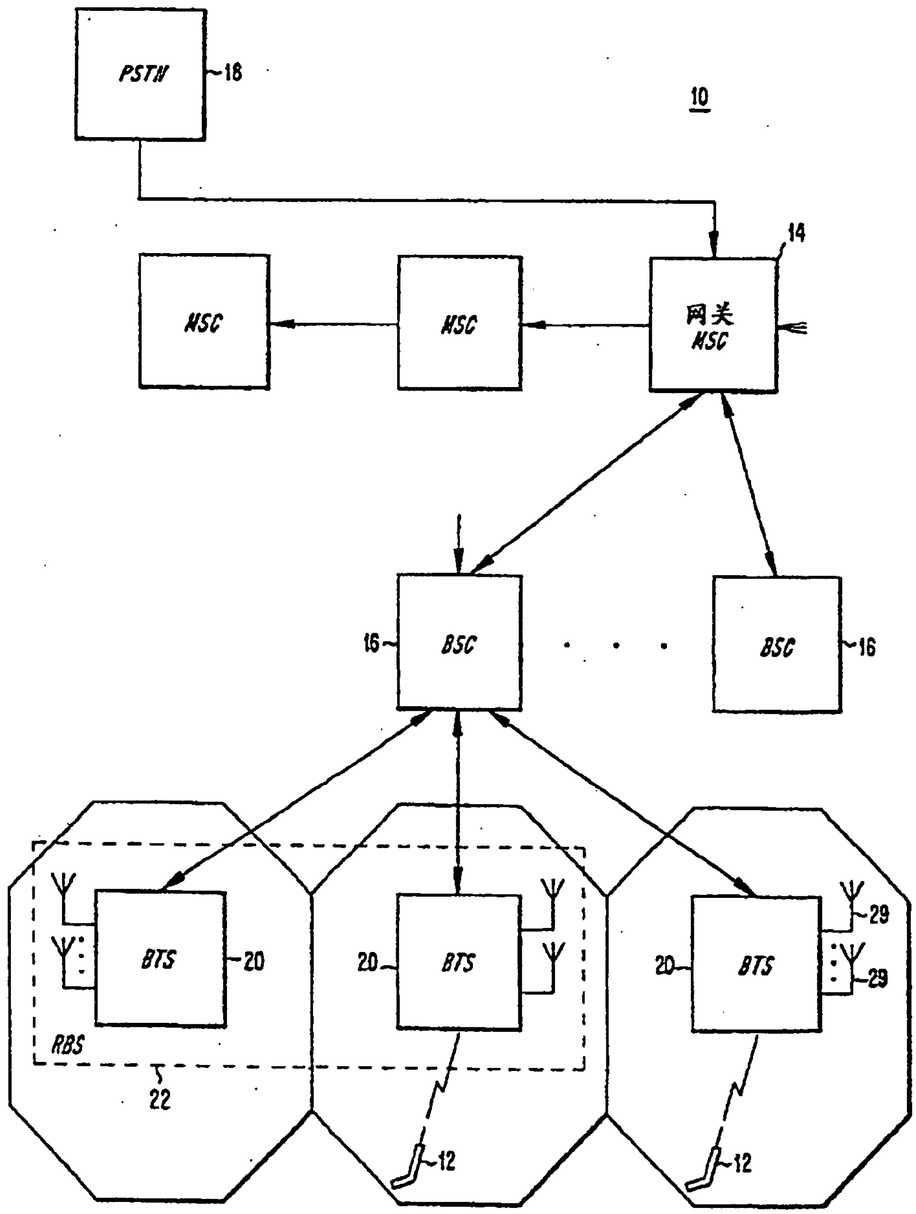

PSTN
18
10
14
网关
MSC
MSC
MSC
16
BSC
BSC
16
BTS
20
20
BTS
20
BTS
29
29
RBS
22
12
12

图 3(a)
(现有技术)

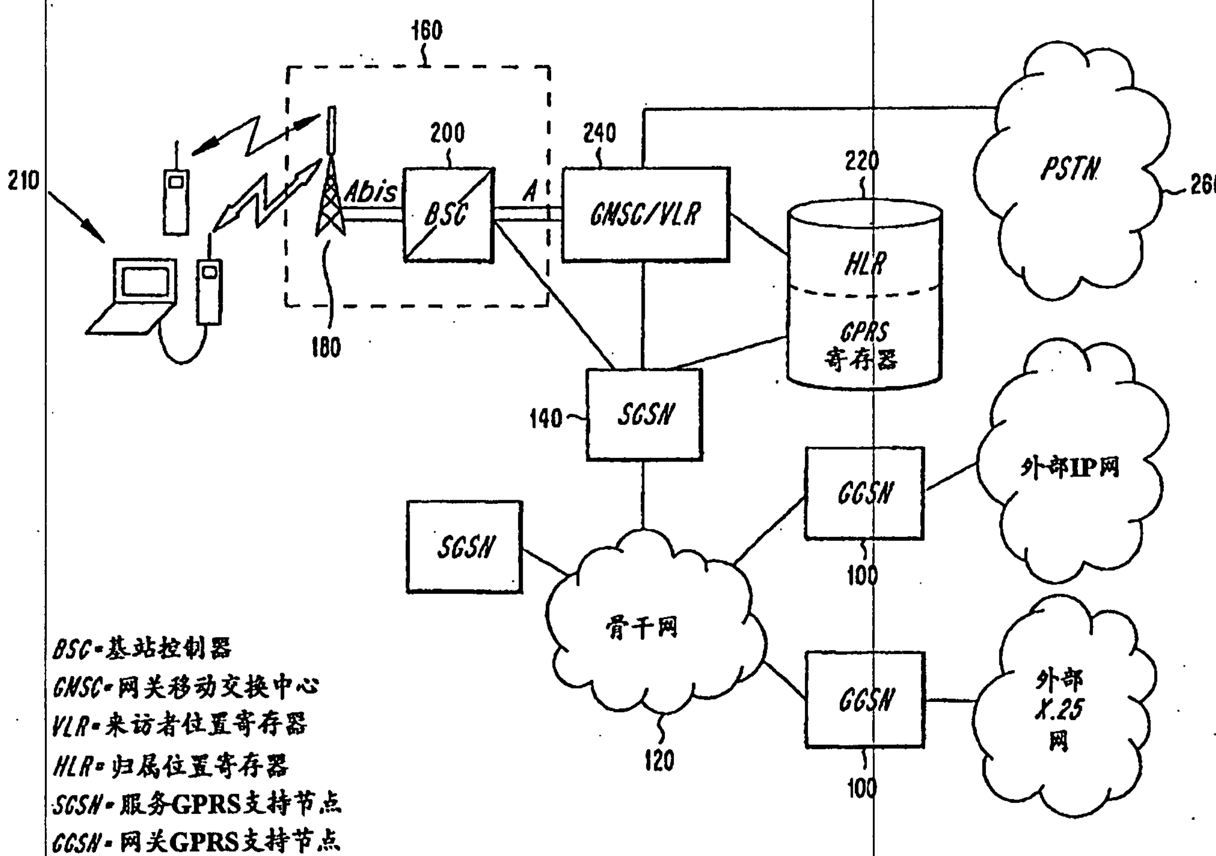

160
210
200
240
220
260
Abis
A
BSC
GMSC/VLR
HLR
GPRS
寄存器
PSTN
180
140
SGSN
SGSN
GGSN
GGSN
100
100
外部IP网
外部
X.25
网
骨干网
120
BSC=基站控制器
GMSC=网关移动交换中心
VLR=来访者位置寄存器
HLR=归属位置寄存器
SGSN=服务GPRS支持节点
GGSN=网关GPRS支持节点

图 3(b)

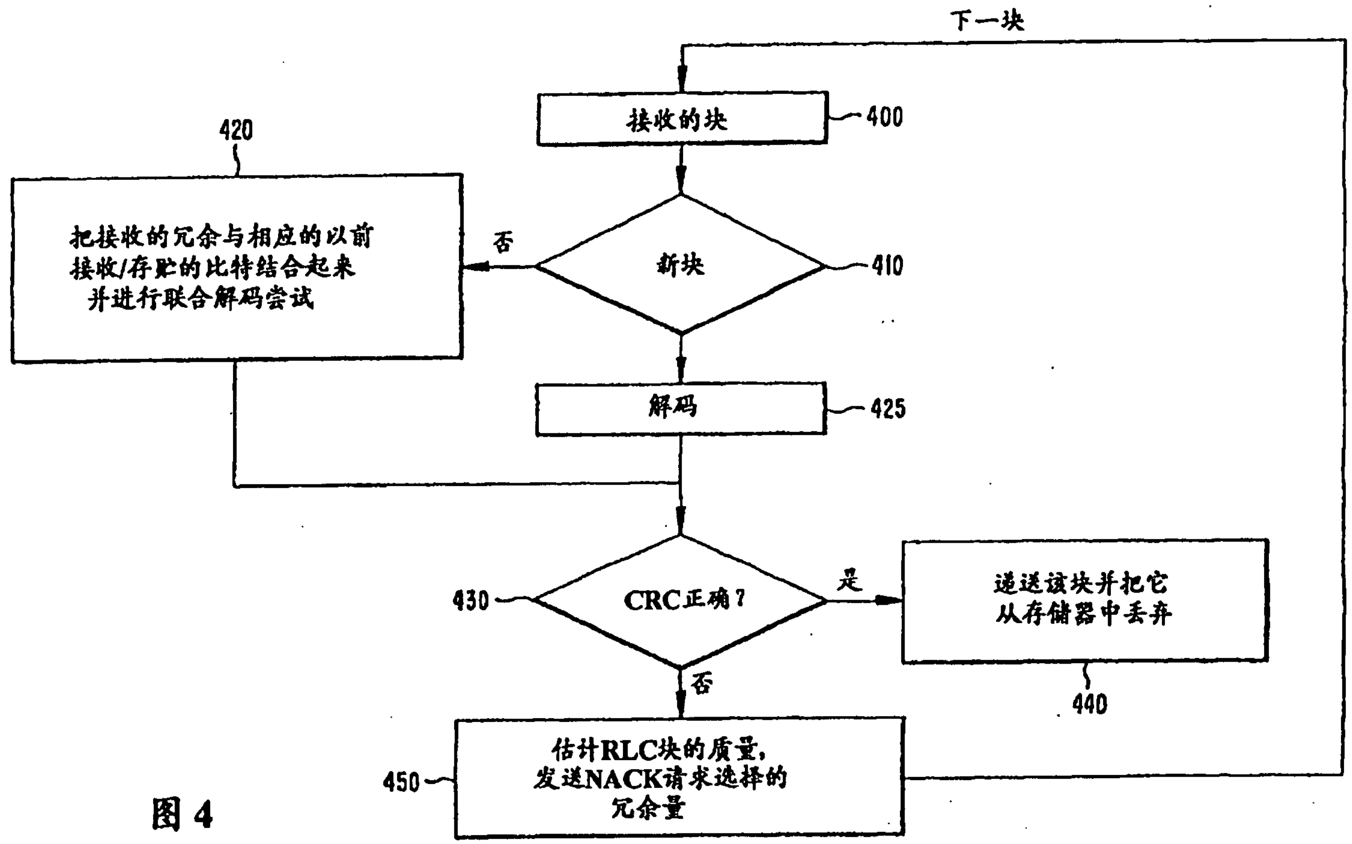

下一块
接收的块
400
420
把接收的冗余与相应的以前
接收/存贮的比特结合起来
并进行联合解码尝试
否
新块
410
解码
425
430
CRC正确？
是
递送该块并把它
从存储器中丢弃
否
440
450
估计RLC块的质量，
发送NACK请求选择的
冗余量

图 4

冗余单元数	编码率	比特组合
0	1	000
1	4/5	001
2	2/3	010
3	4/7	011
4	1/2	100
5	4/9	101
6	2/5	110
7	4/11	-
8	1/3	111

图 5

SSN-5	011	000	101	111	000	000	001	000

图 6

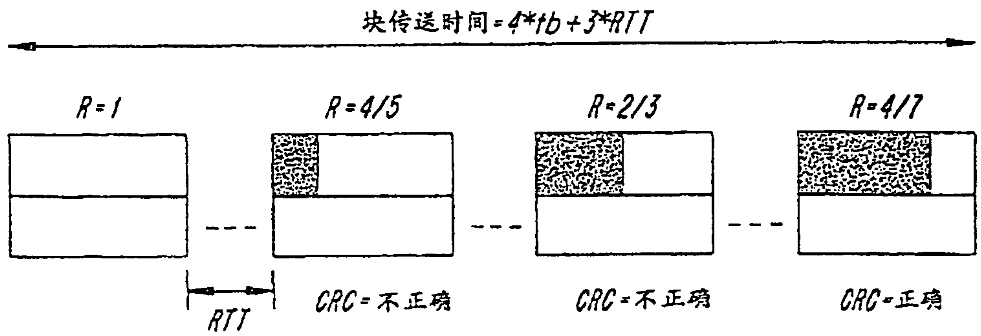

块传送时间=4*tb+3*RTT
R=1
R=4/5
R=2/3
R=4/7
RTT
CRC=不正确
CRC=不正确
CRC=正确

图 7(a)
(现有技术)

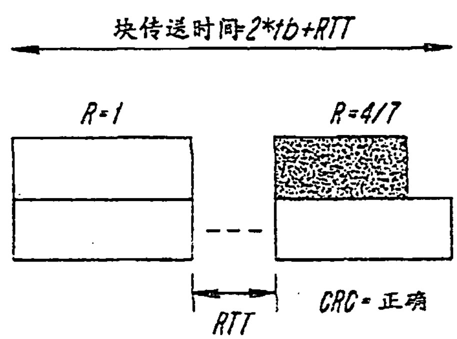

块传送时间=2*tb+RTT
R=1
R=4/7
RTT
CRC=正确

图 7(b)

01.06.08

冗余单元数	编码率	时延(混合 II/III ARQ)	时延(MB 混合 ARQ)
0	1	20 ms	20 ms
1	4/5	240 ms	240 ms
2	2/3	460 ms	240 ms
3	4/7	680 ms	240 ms
4	1/2	900 ms	240 ms
5	4/9	1120 ms	260 ms
6	2/5	1340 ms	260 ms
7	4/11	1560 ms	260 ms
8	1/3	1780 ms	260 ms

图 8

Data communication method and system using an adaptive hybrid-ARQ scheme

Description of corresponding document: US2001056560

BACKGROUND

[0001] The present invention generally relates to error handling in the field of communication systems and, more particularly, to error handling using automatic retransmission requests (ARQ) and variable redundancy in digital communication systems.

[0002] The growth of commercial communication systems and, in particular, the explosive growth of cellular radiotelephone systems, have compelled system designers to search for ways to increase system capacity without reducing communication quality beyond consumer tolerance thresholds. One technique to achieve these objectives involved changing from systems wherein analog modulation was used to impress data onto a carrier wave, to systems wherein digital modulation was used to impress the data on carrier waves.

[0003] In wireless digital communication systems, standardized air interfaces specify most of the system parameters, including modulation type, burst format, communication protocol, etc. For example, the European Telecommunication Standard Institute (ETSI) has specified a Global System for Mobile Communications (GSM) standard that uses time division multiple access (TDMA) to communicate control, voice and data information over radio frequency (RF) physical channels or links using a Gaussian Minimum Shift Keying (GMSK) modulation scheme at a symbol rate of 271 ksps. In the U.S., the Telecommunication Industry Association (TIA) has published a number of Interim Standards, such as IS-54 and IS-136, that define various versions of digital advanced mobile phone service (D-AMPS), a TDMA system that uses a differential quadrature phase shift keying (DQPSK) modulation scheme for communicating data over RF links.

[0004] TDMA systems subdivide the available frequency into one or more RF channels. The RF channels are further divided into a number of physical channels corresponding to timeslots in TDMA frames. Logical channels are formed of one or several physical channels where modulation and coding is specified. In these systems, the mobile stations communicate with a plurality of scattered base stations by transmitting and receiving bursts of digital information over uplink and downlink RF channels.

[0005] Digital communication systems employ various techniques to handle erroneously received information. Generally speaking, these techniques include those which aid a receiver to correct the erroneously received information, e.g., forward error correction (FEC) techniques, and those which enable the erroneously received information to be retransmitted to the receiver, e.g., automatic retransmission request (ARQ) techniques. FEC techniques include, for example, convolutional or block coding of the data prior to modulation. FEC coding involves representing a certain number of data bits using a certain (greater) number of code bits, thereby adding redundancy which permits correction of certain errors. Thus, it is common to refer to convolutional codes by their code rates, e.g., [1/2] and [1/3], wherein the lower code rates provide greater error protection but lower user bit rates for a given channel bit rate.

[0006] ARQ techniques involve analyzing received blocks of data for errors and requesting retransmission of blocks which contain errors. Consider, for example, the block mapping example illustrated in FIG. 1 for a radiocommunication system operating in accordance with the Generalized Packet Radio Service (GPRS) optimization which has been proposed as a packet data service for GSM. Therein, a logical link control (LLC) frame containing a frame header (FH), a payload of information and a frame check sequence (FCS) is mapped into a plurality of radio link control (RLC) blocks, each of which include a block header (BH), information field, and block check sequence (BCS), which can be used by a receiver to check for errors in the information field. The RLC blocks are further mapped into physical layer bursts, i.e., the radio signals which have been GMSK modulated onto the carrier wave for transmission. In this example, the information contained in each RLC block can be interleaved over four bursts (timeslots) for transmission.
[0007] When processed by a receiver, e.g., a receiver in a mobile radio telephone, each RLC block can, after demodulation, be evaluated for errors using the block check sequence and well known cyclic redundancy check techniques. If there are errors, then a request is sent back to the transmitting entity, e.g., a base station in a radiocommunication system, denoting the block to be resent using predefined ARQ protocols.
[0008] Strengths and weaknesses of these two error control schemes can be balanced by combining FEC and ARQ techniques. Such combined techniques, commonly referred to as hybrid ARQ techniques, permits correction of some received errors using the FEC coding at the receiver, with other errors requiring retransmission. Proper selection of FEC coding schemes with ARQ protocols thus results in a hybrid ARQ technique having greater reliability than a system employing a purely FEC coding scheme with greater throughput than a system employing a purely ARQ-type error handling mechanism.
[0009] An example of a hybrid ARQ scheme can be found in GPRS. The GPRS optimization provides four FEC coding schemes (three convolutional codes of different rate and one uncoded mode). After one of the four coding schemes is selected for a current LLC frame, segmentation of this frame to RLC blocks is performed. If an RLC block is found to be erroneous at the receiver (i.e., it has errors which cannot be corrected) and needs to be retransmitted, the originally selected FEC coding scheme is used for retransmission, i.e., this system employs fixed redundancy for retransmission purposes. The retransmitted block may be combined with the earlier transmitted version in a process commonly referred to as soft combining in an attempt to successfully decode the transmitted data.
[0010] Another proposed hybrid ARQ scheme, sometimes referred to as incremental redundancy or type-I hybrid ARQ, provides for additional redundant bits to be transmitted if the originally transmitted block cannot be decoded. This scheme is conceptually illustrated in FIG. 2. Therein, three decoding attempts are made by the receiver. First, the receiver attempts to decode the originally received data block (with or without redundancy). Upon failure, the receiver then receives additional redundant bits R1, which it uses in conjunction with the originally transmitted data block to attempt decoding. As a third step, the receiver obtains another block of redundant information R2, which it uses in conjunction with the originally received data block and the block of redundant bits R1 to attempt decoding for a third time. This process can be repeated until successful decoding is achieved.
[0011] One problem with the technique illustrated in FIG. 2 is the large memory requirement associated with storing the data block (and possibly additional blocks

of redundant bits) until a successful decode occurs, which storage is needed since the subsequently transmitted redundancy blocks (e.g., R1 and R2) are not independently decodable. The storage requirements are multiplied by the fact that the receiver typically stores a multi-bit soft value associated with each received bit, the soft values indicating a confidence level associated with the decoding of the received bit. This problem can be partially solved by employing the technique described in the article entitled "Complementary Punctured Convolutional (CPC) Codes and their Applications" to Samir Kallel, published in IEEE Transactions on Communications, Vol. 43, No. 6, pp. 2005-2009 in June 1995. Therein, the author describes an error correction technique wherein each retransmitted block is itself independently decodable so that when memory space is unavailable previously transmitted blocks can be discarded.

[0012] A second problem encountered with the scheme of FIG. 2 is the large packet transfer delays. These large delays are introduced because, on average, several redundancy retransmissions are required before successful decoding occurs. A third problem associated with the proposed schemes is the inefficient bandwidth utilization due to a stalled ARQ window. The ARQ window is stalled because of the large number of outstanding blocks (i.e., unacknowledged blocks) at a given time.

[0013] Accordingly, it would be desirable to provide new techniques for improving ARQ schemes which reduce overhead signaling, improve the efficiency of memory utilization and minimize the number of redundancy transmissions associated with each decoding in a manner which will permit more efficient processing.

SUMMARY

[0014] These and other drawbacks and limitations of conventional methods and systems for communicating information are overcome according to the present invention, wherein the receiver processes a received block. If the decoding is unsuccessful, a quality estimate is made on the received information. The quality estimate can be based solely on the quality of the particular block which has been erroneously received, solely based on historical data associated with channel quality or it can be some combination of the two. The quality estimate can, for example, be extracted from the soft values that are derived in the receiver. Then, based on the quality estimate, the amount of redundancy required for the successful decoding of the information block is determined. The receiver then sends a not acknowledged (NACK) message to the transmitter identifying the block to be retransmitted along with the amount of desired redundancy, whereupon the desired amount of redundancy is transmitted.

[0015] If the decoding is unsuccessful after the second attempt, then the process continues by determining a second quality estimate associated with both the originally transmitted block and the subsequently transmitted redundant bits. This second quality estimate is then used to determine a next amount of redundant information to be requested, and so on.

[0016] Measurement-based hybrid ARQ schemes according to the present invention will minimize the number of redundancy transmission steps thus reducing the packet transmission delays and the amount of memory required. This is achieved due to the reduced number of ACK/NACK loops required for successful decoding with the measurement based scheme. An exemplary implementation of the present invention provides an estimation of the amount of redundancy depending upon the quality of the received previous data block/redundancy block and/or the quality of the channel. However, other exemplary embodiments of the

present invention also include cases where the amount of redundancy transmitted depends upon other factors such as the amount of memory available, data delay requirements and/or the number of outstanding (unacknowledged) blocks for a given transmission. For example, when the amount of memory is limited or delay requirements are stringent, the redundancy estimation could be scaled up in order to increase the probability of successful decoding in the next step.

BRIEF DESCRIPTION OF THE DRAWINGS

[0017] These and other objects, features and advantages of the present invention will become more apparent upon reading from the following detailed description, taken in conjunction with the accompanying drawings, wherein:

[0018] FIG. 1 depicts information mapping in a conventional system operating in accordance with GSM;

[0019] FIG. 2 illustrates a conventional variable redundancy technique;

[0020] FIG. 3(a) is a block diagram of a GSM communication system which advantageously uses the present invention;

[0021] FIG. 3(b) is a block diagram used to describe an exemplary GPRS optimization for the GSM system of FIG. 3(a);

[0022] FIG. 4 is a flowchart illustrating a measurement-based ARQ scheme according to an exemplary embodiment of the present invention;

[0023] FIG. 5 shows a table describing an exemplary relationship between a number of redundancy units to be transmitted, a coding rate and a corresponding code;

[0024] FIG. 6 shows a format for a an ACK/NACK according to an exemplary embodiment of the present invention;

[0025] FIG. 7(a) illustrates block transmission time using a conventional incremental redundancy scheme;

[0026] FIG. 7(b) depicts an illustrative block transmission time for the same data as in FIG. 7(a) using techniques according to the present invention; and

[0027] FIG. 8 is a table illustrating the cumulative improvement in delay times using the present invention.

DETAILED DESCRIPTION

[0028] The following exemplary embodiments are provided in the context of TDMA radiocommunication systems. However, those skilled in the art will appreciate that this access methodology is merely used for the purposes of illustration and that the present invention is readily applicable to all types of access methodologies including frequency division multiple access (FDMA), TDMA, code division multiple access (CDMA) and hybrids thereof.

[0029] Moreover, operation in accordance with GSM communication systems is described in European Telecommunication Standard Institute (ETSI) documents ETS 300 573, ETS 300 574 and ETS 300 578, which are hereby incorporated by reference. Therefore, the operation of the GSM system in conjunction with the proposed GPRS optimization for packet data (hereafter referred to simply as "GPRS") is only described herein to the extent necessary for understanding the present invention. Although, the present invention is described in terms of exemplary embodiments in an enhanced GPRS system, those skilled in the art will appreciate that the present invention could be used in a wide variety of other digital communication systems, such as those based on wideband CDMA or wireless

ATM, etc.

[0030] Referring to FIG. 3(a), a communication system 10 according to an exemplary GSM embodiment of the present invention is depicted. The system 10 is designed as a hierarchical network with multiple levels for managing calls. Using a set of uplink and downlink frequencies, mobile stations 12 operating within the system 10 participate in calls using time slots allocated to them on these frequencies. At an upper hierarchical level, a group of Mobile Switching Centers (MSCs) 14 are responsible for the routing of calls from an originator to a destination. In particular, these entities are responsible for setup, control and termination of calls. One of the MSCs 14, known as the gateway MSC, handles communication with a Public Switched Telephone Network (PSTN) 18, or other public and private networks.

[0031] At a lower hierarchical level, each of the MSCs 14 are connected to a group of base station controllers (BSCs) 16. Under the GSM standard, the BSC 16 communicates with a MSC 14 under a standard interface known as the A-interface, which is based on the Mobile Application Part of CCITT Signaling System No. 7.

[0032] At a still lower hierarchical level, each of the BSCs 16 controls a group of base transceiver stations (BTSs) 20. Each BTS 20 includes a number of TRXs (not shown) that use the uplink and downlink RF channels to serve a particular common geographical area, such as one or more communication cells 21. The BTSs 20 primarily provide the RF links for the transmission and reception of data bursts to and from the mobile stations 12 within their designated cell. When used to convey packet data, these channels are frequently referred to as packet data channels (PDCHs). In an exemplary embodiment, a number of BTSs 20 are incorporated into a radio base station (RBS) 22. The RBS 22 may be, for example, configured according to a family of RBS-2000 products, which products are offered by Telefonaktiebolaget LM Ericsson, the assignee of the present invention. For more details regarding exemplary mobile station 12 and RBS 22 implementations, the interested reader is referred to U.S. patent application Ser. No. 08/921,319, entitled "A Link Adaptation Method For Links using Modulation Schemes That Have Different Symbol Rates", to Magnus Frodigh et al., the disclosure of which is expressly incorporated here by reference.

[0033] An advantage of introducing a packet data protocol in cellular systems is the ability to support high data rate transmissions and at the same time achieve a flexibility and efficient utilization of the radio frequency bandwidth over the radio interface. The concept of GPRS is designed for so-called "multislot operations" where a single user is allowed to occupy more than one transmission resource simultaneously.

[0034] An overview of the GPRS network architecture is illustrated in FIG. 3(b). Since GPRS is an optimization of GSM, many of the network nodes/entities are similar to those described above with respect to FIG. 3(a). Information packets from external networks will enter the GPRS network at a GGSN (Gateway GPRS Service Node) 100. The packet is then routed from the GGSN via a backbone network, 120, to a SGSN (Serving GPRS Support Node) 140, that is serving the area in which the addressed GPRS mobile resides. From the SGSN 140 the packets are routed to the correct BSS (Base Station System) 160, in a dedicated GPRS transmission. The BSS includes a plurality of base transceiver stations (BTS), only one of which, BTS 180, is shown and a base station controller (BSC) 200. The interface between the BTSs and the BSCs are referred to as the A-bis interface. The BSC is a GSM specific denotation and for other exemplary systems the term Radio Network Control (RNC) is used for a node having similar functionality as that of a

BSC. Packets are then transmitted by the BTS 180 over the air interface to a remote unit 210 using a selected information transmission rate.

[0035] A GPRS register will hold all GPRS subscription data. The GPRS register may, or may not, be integrated with the HLR (Home Location Register) 220 of the GSM system. Subscriber data may be interchanged between the SGSN and the MSC/VLR 240 to ensure service interaction, such as restricted roaming. As mentioned above, the access network interface between the BSC 200 and MSC/VLR 240 is a standard interface known as the A-interface, which is based on the Mobile Application Part of CCITT Signaling System No. 7. The MSC/VLR 240 also provides access to the landline system via PSTN 260.

[0036] Retransmission techniques can be provided in system 10 so that a receiving entity (RBS 180 or MS 210) can request redundant bits associated with an RLC block from a transmitting entity (MS 210 or RBS 180). According to exemplary embodiments of the present invention, the amount of redundant information requested by the receiving entity and transmitted in response to the request (e.g., a not acknowledged (NACK) message) is variable.

[0037] More specifically, the receiver can evaluate the erroneously received RLC block to obtain some estimate regarding how poorly it was received, i.e., its quality. This estimate could, for example, be a measure of bit error rate (BER) or carrier-to-interference ratio (C/I). The receiver then determines the amount of redundancy to request from the transmitter based on the quality estimate for a particular, erroneously received RLC block. In the following discussion, the return of redundancy information is described in terms of redundancy units which can, of course, be any size, e.g., a block of bits, a byte or even a single bit and can be generated in a known manner using a polynomial generator. Generally speaking, the lower the quality estimate, the greater the number of redundancy units that are requested. In addition to (or as an alternative to) basing the amount of redundancy requested on the quality estimate of the erroneously received block itself, systems and methods for error handling according to exemplary embodiments of the present invention may also take into account channel quality over which the block was transmitted and over which the requested redundancy units will be transmitted. For example, the number of redundancy units requested may be based on a global quality measure such as Q=[alpha] * channel quality+(1-[alpha]) * received block quality, where [alpha] is a desired weighting value.

[0038] Thus, an exemplary method according to the present invention is illustrated by the flow chart of FIG. 4. Therein, at block 400, the receiver receives an RLC block which is either data, previously requested redundant bits or some combination thereof. If the RLC block contains only redundant bits associated with a previously received RLC block, then the process moves along the "NO" arrow from decision block 410 to block 420, wherein the redundant bits are combined with previously received/stored bits of a corresponding RLC block and a joint decoding attempt is made. For a more detailed discussion of how redundant bits are matched with earlier received data for joint decoding, the interested reader is referred to U.S. patent application Ser. No. 09/131,166, entitled "Method and System for Block Addressing in a Packet Data Radiocommunication System", filed on Aug. 7, 1998 to Farooq Khan et al., the disclosure of which is expressly incorporated here by reference. Otherwise, if the received block is a new RLC block, the process moves along the "YES" path from decision block 410 to block 425 where the new block is decoded. Then the flow moves to block 430 where a cyclic redundancy check (CRC) is performed. If the CRC passes, i.e., the data is received correctly, then the process moves to block 440 wherein the block is

delivered for subsequent processing, e.g., speech decoding, etc. If the CRC fails, then the flow moves to block 450 wherein an estimate of the quality of the erroneously received block is made, e.g., based on a relative BER or C/I parameter. The quality estimate (and, possibly, other factors described below) is then used to select a desired amount of redundant bits to be used in the next decoding attempt. The receiver then transmits an NACK message associated with this (and possibly other) RLC blocks, which NACK message indicates the amount of redundancy that the receiver wishes for the transmitter to send. The flow then loops back to process the next received block.

[0039] Those skilled in the art will appreciate that requesting the number of redundancy units to be transmitted can, in exemplary embodiments employing convolutional encoding, be considered as substantially equivalent to specifying a desired coding rate for a particular block that was erroneously received. For example, as illustrated in the table of FIG. 5, for an RLC block containing four "units" of data, requesting any number from 1-8 of redundancy units to be transmitted results in a different effective coding rate for the second attempt at decoding the data. Thus, for example, an erroneously received RLC block that has nonetheless relatively high quality, may result in the receiver requesting only one redundancy unit from the transmitter. A very poorly received RLC block may, on the other hand, result in the receiver requesting eight redundancy units for that specific RLC block. The particular relationship between estimated RLC block quality and number of redundancy units requested may vary from system to system and can, for example, be optimized through simulation to achieve the desired result of minimizing the number of decoding attempts per block as described below.

[0040] Once the receiver has evaluated the quality of received RLC block and selected a desired amount of redundancy, it will include this information in a report to the transmitter. Using the example of FIG. 5, each different number of redundancy units which can be transmitted may be assigned a different code or bit combination. Then, the receiver can send an acknowledged/not acknowledged (ACK/NACK) message identifying the amount of desired redundancy, if any, for each recently received RLC block. An example is provided in FIG. 6.

[0041] Therein an ACK/NACK message containing the information [(5(3), 6(0), 7 (5), 0(8), 1(0), 2(0), 3(1), 4(0)] is illustrated. In the foregoing notation, "5(3)" denotes that three redundancy units are requested by the receiver for the RLC block having a sequence number of 5. For the RLC block having a sequence number of 6, the receiver has included the code 000, indicating that no redundancy information need be transmitted since that RLC block was correctly received.

[0042] As mentioned earlier, by tailoring the amount of redundancy requested to the quality of the received block, Applicants anticipate that fewer decoding passes will be needed per block as compared with conventional techniques wherein the same amount of redundancy is transmitted for each erroneously received block. This point will be more evident upon consideration of FIGS.7(a), 7(b) and 8.

[0043] Therein, exemplary block transfer times using the conventional incremental redundancy scheme (FIG. 7(a)) and the measurement-based variable redundancy scheme are compared. For this purely illustrative example, a block period equals 20 ms, a round trip time (RTT) between transmission of an RLC block by a transmitting entity and receipt of a corresponding ACK/NACK message by that transmitting entity is 200 ms and an erroneously received RLC block needs three units of redundant information (i.e., a coding rate of 4/7) to be properly decoded. Thus, in FIG. 7(a) it will be seen that four transmissions are required until the CRC passes for this RLC block, wherein after each failure the transmitting entity sends

an additional unit of redundancy. By way of contrast, employing the present invention, the receiver is able to request three units of redundancy based on the estimated quality of this RBC block so that only two passes are needed, thereby reducing the block transfer delay from 680 ms to 240 ms, respectively. Those skilled in the art will appreciate that the actual different in delays associated with the two techniques may also vary depending upon other conditions, e.g., varying radio channel conditions. Moreover, the delay difference will increase with the number of redundancy transmission steps used in the conventional technique as illustrated by the table in FIG. 8. It will be understood that the numerical values provided in the foregoing example are merely illustrative and intended to make clearer advantages associated with the present invention.

[0044] In addition to reducing delay, exemplary embodiments of the present invention also reduce the likelihood that the ARQ window will stall and reduce memory requirements. This is because techniques according to the present invention minimize the number of outstanding blocks by ensuring a faster block decoding and delivery. By preventing a stalled ARQ window condition, more efficient bandwidth utilization is obtained since new RLC blocks cannot be transmitted during a stalled condition.

[0045] As mentioned earlier, the block of data as originally transmitted may include some redundant information, i.e., may have some level of FEC coding. This initial level of FEC coding may be determined by the transmitting entity based upon information that the transmitting entity receives regarding the channel quality. For example, a mobile station may make estimates regarding channel quality and forward those estimates to a base station. Then, the base station can use the received channel estimates to select an appropriate amount of redundancy to transmit with the payload information to the mobile station. Preferably, the base station would select an amount of redundancy which will allow the mobile station to decode the data block on its first attempt given the channel quality estimate. However, those skilled in the art will appreciate that the base station may select a greater or lesser amount of redundancy depending upon various current system factors such as those described earlier.

[0046] Although the invention has been described in detail with reference only to a few exemplary embodiments, those skilled in the art will appreciate that various modifications can be made without departing from the invention. For example, information regarding the number of redundancy units to be transmitted could be passed back to the transmitter implicitly, e.g., by sending the estimated quality of measures for each block, rather than explicitly as in the example of FIG. 6. The transmitter will then determine an appropriate number of redundancy units to return. In determining the number of redundancy units, the transmitter may take into account, in addition to the received quality measures, other factors such as resource availability, etc. Accordingly, the invention is defined only by the following claims which are intended to embrace all equivalents thereof.

Data communication method and system using an adaptive hybrid-ARQ scheme

Claims of corresponding document: **US2001056560**

What is claimed is:

1. A method for transferring information over a communication link comprising the steps of:
receiving a block of data over said communication link;
determining a quality level of at least one of said received data block and said communication link; and
requesting a quantity of additional information associated with said data block, said quantity selected based on said determined quality level.

2. The method of
claim 1, wherein said step of determining further comprises the step of:
estimating a bit error rate associated with said received data block.

3. The method of
claim 1, wherein said step of determining further comprises the step of:
using soft information obtained during a decoding process to determine said quality level.

4. The method of
claim 1, wherein said step of requesting further comprises the step of:
transmitting a message identifying said data block and said quantity of additional information.

5. The method of
claim 1, further comprising the step of:
selecting, as said quantity of additional information, a number of units of redundant information.

6. The method of
claim 5, wherein said selected number of units of redundant information varies inversely relative to said determined quality level.

7. The method of
claim 1, wherein said step of determining said quality level further comprises the step of:
determining said quality level based solely on said received data block.

8. The method of
claim 1, wherein said step of determining said quality level further comprises the step of:
determining said quality level based solely on a quality of said communication link.

9. The method of
claim 1, wherein said step of determining said quality level further comprises the step of:
determining said quality level based on a combination of quality information associated with said received block and quality information associated with said communication link.

10. The method of
claim 1, further comprising the step of:
transmitting said requested quantity of additional information.

11. The method of
claim 1, further comprising the step of:
transmitting a quantity of additional information which is different than said requested quantity.

12. The method of
claim 11, wherein said transmitted quantity of additional information and said requested quantity of additional information differ based on at least one of: a memory usage parameter, a number of outstanding blocks and a resource availability.

13. An apparatus for transferring information over a communication link comprising:
means for receiving a block of data over said communication link;
means for determining a quality level of at least one of said received data block and said communication link; and
means for requesting a quantity of additional information associated with said data block, said quantity selected based on said determined quality level.

14. The apparatus of
claim 13, wherein said means for determining further comprise:
means for estimating a bit error rate associated with said received data block.

15. The apparatus of
claim 13, wherein said means for determining further comprises:
means for using soft information obtained during a decoding process to determine said quality level.

16. The apparatus of
claim 13, wherein said means for requesting further comprises:
means for transmitting a message identifying said data block and said quantity of additional information.

17. The apparatus of
claim 13, further comprising:
means for selecting, as said quantity of additional information, a number of units of redundant information.

18. The apparatus of
claim 13, wherein said selected number of units of redundant information varies

anduandbandd

inversely relative to said determined quality level.

19. The apparatus of
claim 13, wherein said means for determining said quality level further comprises:
means for determining said quality level based solely on said received data block.

20. The apparatus of
claim 13, wherein said means for determining said quality level further comprises:
means for determining said quality level based solely on a quality of said communication link.

21. The apparatus of
claim 13, wherein said means for determining said quality level further comprises:
means for determining said quality level based on a combination of quality information associated with said received block and quality information associated with said communication link.

22. The apparatus of
claim 13, further comprising:
means for transmitting said requested quantity of additional information.

23. The apparatus of
claim 13, further comprising:
means for transmitting a quantity of additional information which is different than said requested quantity.

24. The apparatus of
claim 23, wherein said transmitted quantity of additional information and said requested quantity of additional information differ based on at least one of: a memory usage parameter, a number of outstanding blocks and a resource availability.

25. A method for decoding information blocks in a radiocommunication system comprising the steps of:
receiving a block of information;
decoding said block;
performing an error detection technique on said decoded sblock;
if said block is determined to have been erroneously received, then determining a quality level;
selecting, based on said quality level, a desired amount of redundant information;
transmitting a request for said desired amount of redundant information to a transmitting entity;
receiving said requested amount of redundant information; and
jointly decoding said block of information and said redundant information.

26. The method of
claim 25, wherein said step of performing further comprises the step of:
performing a cyclic redundancy check (CRC) on said block of information.

27. The method of
claim 25, wherein said quality level is a quality of said received block.

28. The method of
claim 25, wherein said quality level is a quality of a channel over which said redundant information will be transmitted.

29. A method for communicating information between a transmitting entity and a receiving entity comprising the steps of:
estimating, at said receiving entity, a channel quality;
transmitting, by said receiving entity, an indication associated with said channel quality; and
transmitting, by said transmitting entity, a block of information plus an amount of redundancy associated with said information, wherein said amount is based on said indication.

30. A method for transferring information between a first transceiver and a second transceiver comprising the steps of:
receiving a data block at said first transceiver;
estimating a quality associated with one of said received blocks and a channel;
transmitting by said first transceiver, said estimated quality to a second transceiver;
determining, at said second transceiver, an amount of redundancy based, at least in part, on said estimated quality; and
transmitting, by said second transceiver, said amount of redundancy to said first transceiver.

31. A method for communicating information between a first transceiver and a second transceiver comprising the steps of:
receiving a data block at said first transceiver;
estimating a first quality associated with at least one of said received block and a channel;
transmitting, by said first transceiver, an indication associated with a selected amount of redundancy based on said estimated first quality;
transmitting, by said second transceiver, said selected amount of redundancy to said first transceiver;
estimating, at said first transceiver, a second quality associated with at least one of both said received block and received redundancy information and said channel quality; and
transmitting an indication of said estimated second quality to said second transceiver.

anduandbandd